Chapter 8

Audio Driver

8.1 Why Audio Drivers?

Logic automatically recognizes what audio hardware is available on your computer, and configures itself accordingly. Access to the hardware parameters is carried out using drivers, which are covered in the this chapter.

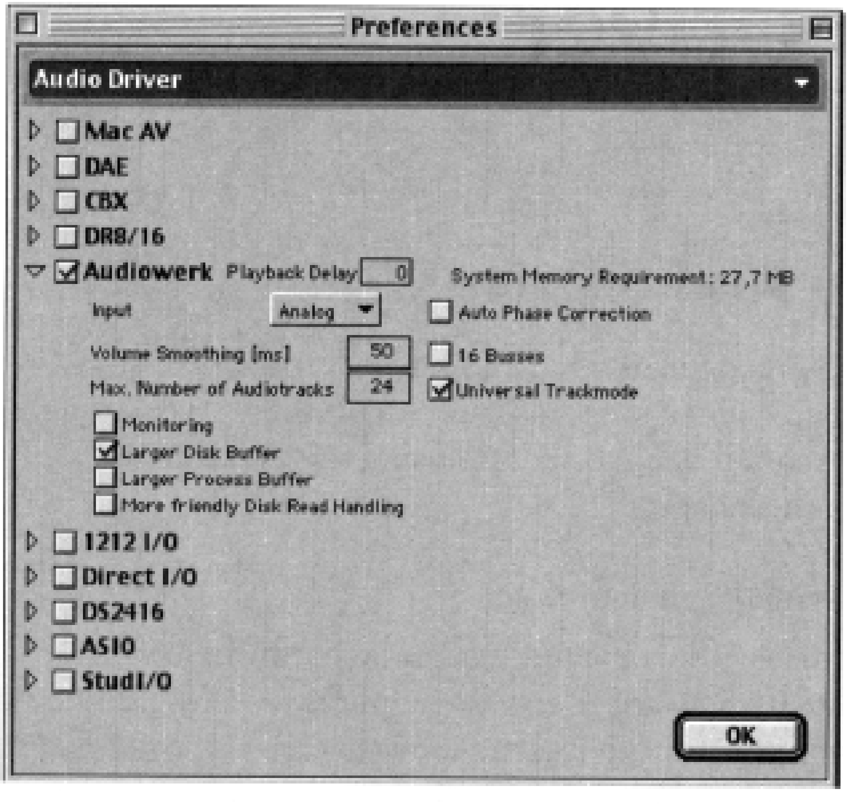
Using Logic Audio Platinum without "Audio"

If you don't want to use a specific piece of audio hardware with Logic (for example, if you want to run Logic purely as a MIDI sequencer, or if you just want to use the internal computer hardware for listening to CDs), select **Audio > Audio Hardware & Driver.** Uncheck the box for the audio hardware you normally use.

You may have to restart Logic before any new settings become effective. When activating a driver (using the **Audio > Hardware & Drivers** option) there is an option in the dialog window ("Try Re-Launch").

> **This process is always only an attempt to start, as it is possible that not enough RAM is available, Making memory available to the driver is simpler when starting Logic Audio.**

If the attempt fails, you should leave Logic and restart. The driver will then be loaded with the program.



Available Drivers

Any Audio hardware requires an appropriate driver as a communication link to the Logic software. Some of the Drivers support several different types of Audio Hardware. Here is a brief list of the Drivers available. These are described in more detail later:

Mac AV

The driver for the AV hardware, i.e. the computer's audio inputs and outputs, is covered in the Installation manual. Please refer to it for the relevant settings in the Sound control panel. The Mac AV driver has several parameters which are described in the section *Common Parameters* on page 8-6.

DAE

DAE stands for *Digidesign Audio Engine.* This program is the driver software for many Digidesign cards. New cards, like "Project II" are not supported by the DAE, but by the driver

"Direct I/O". If you use an older PCI card, such as an Audiomedia III or a ProTools System, please be aware that you may control your system with either DAE, or the Direct I/O driver. You'll find detailed information about DAE in the section ***DAE*** on page 8-9, and more about Direct I/O in the section ***Direct I/O*** on page 8-38. It's not possible to use DAE and Direct I/O simultaneously. The DAE driver for Logic Audio Gold supports all Digidesign audio cards except TDM hardware, with Logic Audio Platinum it also supports TDM hardware.

CBX

Logic supports the CBX D3 and CBX D5 hard disk recorders by Yamaha. You'll find more information about the CBX driver in the section ***CBX*** on page 8-15.

DR 8/16

Logic supports the DR 8 and DR 16 hard disk recorders by Akai. You'll find more information about the DR 8/16 driver in the section ***DR 8/16*** on page 8-17.

Audiowerk

If you want to use Audiowerk8 or Audiowerk2-no problem. It will be recognized by the program, and configured automatically. ***Input*** selects between the digital S/PDIF - and the analog Input. ***Monitoring*** selects whether a monitor signal is output or not. The Audiowerk Driver supports the features of the Audio engine, as described in the section ***Audio Engine*** on page 8-5.

With the Audiowerk2, "Output 1-2" are considered as the analog, and "Output 3-4" as the digital S/PDIF outputs.

1212 I/O

The Korg 1212 I/O is a PCI-Card with 8-Channel ADAT-type I/Os and two-channel analog and S/PDIF I/O. Its parameters are described in the section 1212 I/O on page 8-20.

Direct I/O

This is the driver for newer Digidesign products like Project II, that can't be controlled by DAE. Direct I/O may also be used with other Digidesign-PCI-Cards, like Audiomedia III. In comparison to DAE, there are more tracks, there are native Effects and VST-Plug-ins. You also can control a ProTools III Core System and even ProTools 24 (d24) or ProTools 24 MIX-Systems via Direct I/O. But please keep in mind that using these system with Direct I/O disables all TDM functions. The Direct I/O Driver supports the features of the Audio engine, as described in the section *Audio Engine* on page 8-5. Read more about Direct I/O in the section *Direct* I/O on page *8-38,* and more about DAE in the section *DAE* on page *8-9.* It's not possible to use Direct I/O and DAE simultaneously.

Sonorus StudI/O

This hardware supports 24-Bit recording, high sample rates such as 88,2 or 96 kHz, when used in conjunction with Platinum, and has ADAT and S/PDIF-In/Outs. Read more about this driver in the section Sonorus *StudI/O* on page *8-40*.

ASIO

ASIO stands for "Audio Stream In/Out", a standard introduced by Steinberg for communication between audio software like Logic, and audio hardware. The number of supported devices changes continuously. Please see the ReadMe file on the program CD. It contains hints how to adjust the preferences for each type of audio hardware. The ASIO Driver supports the features of the Audio engine as described in the section *Audio Engine on* page 8-5. You'll find more Information about ASIO in the section *ASIO* on page *8-40.*

Yamaha DSP Factory (DS 2416)

You can read all about the driver for this digital mixing desk hardware in the section *Yamaha DSP Factory* on page *8-43*.

8.2 Audio Engine

The features available vary somewhat from driver to driver. The Audio Engine used for Audiowerk, Mac AV, Korg 1212, Direct I/O, Sonorus StudI/O and ASIO offers the most flexibility. This covers the vast majority of available audio hardware:

16 and 24 bit audio files can be played back simultaneously with Logic Audio Platinum. Playback of 24 bit audio files naturally is only possible with 24 bit hardware, and therefore is not possible with the Audiowerk.

Windows Wave files (*.wav) can be played back directly. They appear in the file selector box, when adding an audio file, provided the file type is “WAVE”, or the file name ends with the extension "*.wav" Important: Please avoid playing back from MS-DOS formatted storage media, as the performance is very poor under Mac OS. Please copy the files to an HFS drive (Mac OS formatted) beforehand.

Level meter for bus and output objects. The level meter displays the signal after the Insert Slots (Plug-ins).

Audio tracks can be routed to busses (in addition to the regular output routing.) This allows you to create sub groups (e.g. all drum tracks) and edit them together, e.g. with filters.

Busses can be routed to other busses. This extends the routing options, e.g. for processing combined sub group signals.

Bouncing to 8, 16 or 24 bit audio files using the SDII AIFF or Wave file formats.

The best fader response and volume smoothing possible.

Best performance possible with internal, or VST plug-ins.

Stereo Recording. Stereo recordings normally create an interleaved stereo file (single file for a stereo recording).

If the files have to be compatible with Digidesign DAE/TDM Systems, split stereo files have to be created. Select **Audio > Audio Preferences > force convert interleaved into split stereo files** for DAE/TDM compatibility.

Common Parameters

The drivers mentioned above not only have the audio engine, but therefore also the following parameters in common:

Volume Smoothing [ms]

This parameter defines the length of the fade between two consecutive volume values for an audio track. When setting this value to 0 you might hear "zipper noise" when moving a volume fader during playback. Higher values soften the volume changes and eliminate the zipper noise.

Max. Number of Audiotracks

The Audio Engine requires free system memory, which is not assigned to Logic or any other application. The amount of memory needed depends on the maximum number of tracks to be played, and on the number of I/O channels supplied by the driver. This setting allows you to reduce the amount of memory used by the driver, by reducing the number of tracks. This may be sensible when you want to run other applications or drivers simultaneously.

20/24 Bit Recording

With Logic Audio Platinum, you can record 24 bit files when this setting is active. Please keep in mind that this only makes sense if you are actually using a 20 or 24 bit interface. As long as your audio hardware is capable of this, you can select this option in the **Audio > Audio Hardware & Drivers** window. 20 or 24 Bit Recordings offer a significant improvements in the available dynamic range, however, they require high quality peripheral components.

96 kHz

Some audio cards allow high sampling rates such as 88.2 or 96 kHz. Logic Audio Platinum supports these high sample rates. The sample rates, as usual, can be selected in the **Audio > Sample Rate.** window. Keep in mind that with these higher sample

rates, not only that twice the space on the hard disk will be needed, but also that the Audio Engine will be required to perform twice as fast. Furthermore, you should consider that the improvement in audio quality of 96kHz recordings compared to 44.1 kHz in comparison with the differences between 16 and 24 bit recordings, is little. Many audio engineers consider 44.1 kHz 24-bit recording to be the best balance of sound quality, and efficient use of resources.

Monitoring

This option allows you to switch monitoring (i.e. listening to the actual input signal) on or off. Please note that monitoring is processed only via software - a certain delay is inevitable (see “Buffer Size”). If you are listening to the recorded signal through your mixing desk, you should switch this option off.

The Korg 12/12 I/O features latency-free monitoring.

Larger Disk Buffer

This option influences the amount of audio data that is read from the disk in advance. Since version 4.0, this option is switched off by default, matching the demands of fast hard drives and powerful computers. If you get frequent error messages while running Logic Audio in this mode, you should switch this setting on, so that you can play back more tracks, achieving higher reliability. However, more RAM is needed in this case.

Larger Process Buffer

This parameter determines the size of the native buffer used to compute mixers and effects. Do not activate this option if you own a fast computer (G3). This shortens response times to operations such as volume changes or Solo.

When you’re working with older PowerPC CPUs, the smaller process buffer may cause problems. In this case, activate the option. This will increase the number of effects you can use simultaneously.

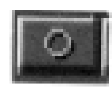
When in doubt - e.g. if you own an older Power Macintosh model but are using a G3 card - experiment. Experiment to find the setting that coaxes the best performance from your system.

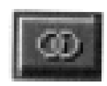
More friendly Disk Read Handling

Activate this setting when using other Audio drivers simultaneously, if you experience playback problems. This gives the other drivers more time for their disk access. The maximum number of audio tracks for this driver may be reduced when this option is enabled.

Universal Track Mode

With **Audio > Audio Hardware & Drivers > Universal Track Mode** engaged, you can play back adjacent stereo and mono regions on a single track. The even-numbered audio objects won't be regarded as the right channels of the odd -numbered stereo audio objects to their left, and every audio object has it's own mono/stereo switch. Depending on whether a mono or stereo region is played back, the pan knob will behave as a balance or pan control. If you play back a mono region and the pan is set to the center position, both channels of the audio object will output the same level. Please note that the Universal Track Mode has limited routing capabilities.





If the Universal Track Mode is not engaged, mono regions will be played back only on the left or right channel of stereo tracks. You can only route the signal to single bus objects. The Non-Universal Track Mode has the advantage of more routing capabilities, such as additional output sends. You can also route the signal to stereo sends and pairs of busses (stereo busses), if you switch Universal Track Mode off.

The Non-Universal Track Mode is useful if you want to play different mono files for left and right mono channels of one audio object, even when it is linked to be a stereo track. An inserted stereo/stereo plug-in on that stereo linked track receives different signals for left and right, which is useful for vocoder like plug-ins.

In order to change tracks to or from DAE/TDM always switch Universal Track Mode off. Also, be aware that DAE/TDM currently does not work with interleaved stereo files, but most other drivers like Audiowerk8 Direct I/O or ASIO and others do. If

your tracks have to be switched between DAE/TDM and others, split stereo files should be used. To do so, enable the following switch: **Audio > Audio Preferences > Global > Force convert *interleaved into split stereo file(s).***

After changing the Universal Track Mode setting, you have to reboot Logic before the change will take effect.

8.3 DAE

With Logic Audio, the Digidesign Audio Engine is currently used to drive the following audio hardware:

AudioMedia II and III
Session 8
SoundTools
Protools II, III, and 24

ProTools Mix

As Digidesign TDM systems also use the DAE (Digidesign Audio Engine), you should read the first two sections of this chapter, if you have a TDM system. For information on the special features of TDM, see the section ***TDM Hardware*** on page 8-22.

Starting Logic Audio without DAE

If you hold down the ctrl key when you start Logic, a dialog box appears, which allows you to switch off individual drivers, or all audio drivers, for this particular start-up. You can start Logic more quickly without Audio drivers, if for example, you want to use just the MIDI sequencer.

Digidesign Hardware

The **Digidesign Hardware Setup** dialog box contains all the important System settings for your Digidesign hardware.

This is where you determine whether Logic will use the analog or digital inputs when recording.

You can also check whether there are any communication problems with your hardware-for example if your hardware is not being recognized properly.

If you own several pieces of Digidesign hardware, you can select from among them here.

Opening the Setup Window

You can open the Setup window from within the Audio window by selecting **Options > Digidesign Hardware Setup.**

Select Card Type

This is where you select the card whose settings you want to adjust. Please note that Pro Tools III hardware can be treated like a Session 8 card, if you wish.

Cards to Use

If you have more than one card of the same type, this is where you select the one whose settings you want to change, by entering the relevant PCI or NuBus slot number from a flip menu, showing all the possible numbers.

Interface Options

This is where you make the settings for a given card.

Interface

Some Digidesign systems offer you a choice of audio interfaces, when you buy them. You set the kind of interface you're using from this box.

Sample Rate

This is where you set the sample rate you will be using in your recordings. All the audio files you record with Logic (or any other software) will be recorded at the rate you enter here. You

can select either 44.1 or 48 kHz This parameter corresponds to the setting in the Audio menu.

Sync Mode

This is where you select the source for the system's clock signal (or bit clock-pulses sent out at the sample rate frequency). You have the following options:

- Internal -

The Digidesign hardware's internal sync signal is used. It runs at the frequency you've selected in the ***Sample Rate*** box above.

The following note applies to ProTools only:
In a TDM system, only one of the cards determines the sync rate; the card in the NuBus or PCI slot with the lowest number. This should be the Disk I/O card, You can tell which is the "Master sync card" by its glowing red LED.

- Digital -

The sync signal received at the digital input is used.

Any digital output, whether AES/EBU or S/PDIF is constantly sending clock information, even if a signal is not passing through it. When this setting is on Digital, the sync signal is used by the Digidesign hardware, not only when recording, but also as a sync clock for the whole system for playback. If this box is set to Digital, please be sure that the digital device connected to the input is never turned off, or you will Lose your system sync signal.

For example, if you are using a portable DAT recorder that saves power by turning itself off when not in use, you should select Sync Mode: Internal immediately after recording from the digital input.

CH 1-2 Input

This is where you choose whether you want to use the analog inputs (for which you select the ***Analog*** setting) or the digital inputs (for which you select ***Digital).***

If you are using an audio interface with more than one pair of digital inputs (e.g. an 888 I/O), you will also be able to select any of the extra input pairs from here.

Swapping Digidesign Hardware

If you have more than one kind of Digidesign hardware installed in your system (for example Pro Tools and Session 8) you can swap between the different types by selecting **Options > Exchange Digidesign Hardware** from the Audio window (provided the Session 8 card is the first card in your computer).

Pro Tools can read audio files from the hard drive attached to the Session 8 card.

Setting a Level for Recording

In a studio, the recording level will normally be set at the bus fader on the mixing desk (or directly at the sound source). The recording machine should be set so that a level of 0 VU (Volume Units) corresponds to the relevant reference level of +6 dBu (radio), +4 dBu (professional studio) or -10 dBV (home recording).

However, if in certain situations, you need to be able to set the recording level at the input of Logic, some HDR hardware will allow you to do so:

Open the Audio window and select **Options > Digidesign Hardware Setup,**

Click **Other Options.**

You can now set the recording level for the AudioMedia card.

On a ProTools system with the 888 I/O audio interface, you can calibrate the recording level, using the “Calibration Tool” software supplied by Digidesign. For example, this is very useful if you want to set the reference level to give you greater headroom.

Session 8 / ProTools Project

The first time you start the program with Session 8 or ProTools Project hardware, and DAE Version 2.95 or later, Logic asks which driver mode should be used: Project or Session 8.

The **Audio > Audio Hardware & Drivers > Use Project Driver Mode for Session 8 System** option means you can swap modes later.

Advantages of Project drivers:

Allows scrubbing of up to 2 audio tracks
I/O routing for every track, directly from the audio objects
Up to 4 sends per track

Disadvantages of Project drivers:

Internal/External mix modes not available
Routing (with Studio Interface) not available
Only 4 EQs available in total (spread over all 8 tracks).

To help you decide:

If you want scrubbing, use Project driver mode.

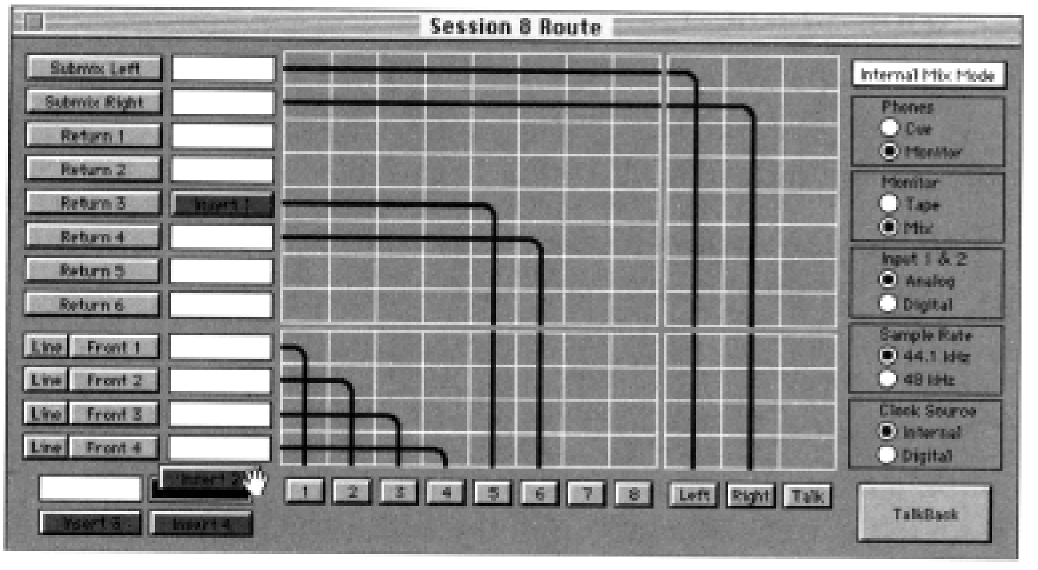
If you want 8 EQs and Session 8 routing, use Session 8 mode.

Plug-in Start Window

Whenever you start TDM Plug-ins, a small status window opens. If there is a problem, this window should show which Plug-in(s) are the cause. If you want to disable this window, you can switch it off, by selecting **Audio > Audio Hardware & Drivers > DAE/TDM > TDM Setup Indicator** (default: on).

Session 8 Routing

You can determine the Session > routing from the audio object.



You use this window to set the various Session 8 routing options. For more on these options, please refer to the Session 8 manual.

Audio Objects with Session 8 in Internal Mix Mode

As with Pro Tools II, additional sends have to be set up individually for each channel. Click on one of the fields under the word Sends. You use the pull-down menu that appears, to set the number of the required send. To the right of this is the control used for setting the aux send level.

Internal Mix Mode allows you to control the stereo master channels. Simply define audio objects with the setting ***Output 1*** for the left-hand channel, and ***Output 2*** for the right. The master channels can be controlled by one stereo fader, if they are set to ***Output 1-2***. The ***Output 3/4*** settings can be used for fader control of the Cue L/R channels, while the ***Output*** 5-8 settings can be used for Aux Master Send controls.

Session 8: Points to Note

The Session *8* ***Mode is available with Logic Audio Gold, but not with Platinum-for technical reasons. Platinum-customers can use the***

Gold version without any disadvantages. Platinum supports TDM system instead.

Overviews

Session 8 hardware cannot calculate, or display any overviews (the Logic graphic files that represent the waveforms you store) while recording. The overviews are created immediately after recording stops. Once you've finished recording, however, the overview calculation is carried out "in the background", so you can engage other functions while the waveform display is being created-you don't have to sit and wait.

Scrubbing

Audio scrubbing is not possible with Session8.

Sampling Rate

Session8 has two fixed sampling rates: 44.1 kHz and 48 kHz. This means that you cannot calibrate it to external timecode, as this function varies the sample rate to match the existing time-code. It's possible that Digidesign will revise this part of the DAE software, to make Session 8 compatible with this function in a future version. As a user, you should remain in close contact with Digidesign, so you can enjoy the benefits of the latest available software updates.

8.4 CBX

Logic Audio Platinum supports the h CBX-D3 and CBX-D5 hard disk recording systems by Yamaha.

CBX-D3

If the CBX-D3 operating system software is not yet installed, LOGIC Audio automatically installs it. While it does this (it

happens after booting up LOGIC Audio), the message ***"Initializing CBX"*** will appear on the screen.

The CBX-D3's sample rate can be calibrated to an external SMPTE signal, or external MTC. Please note that the external clock source should run at the same speed when calibrating, as it will at mixdown The best way to insure this is to calibrate immediately before mixdown.

Using a CBX-D3 and -D5 together

Please give the CBX-D3 the lower SCSI-ID.

In the **Audio > Audio Hardware & Drivers** window, select ***Continous Playback*** under CBX ***Driver Mode.*** If the hard disk then seems too slow, change the setting to ***Mixed Mode.***

Restrictions

Level Metering while Recording

The current version of the CBX operating system software does not allow level metering while recording. This is not the fault of LOGIC Audio. Instead of the correct level, the meters on CBX audio objects will cycle up and down to show activity. Please use the level meters on the CBX itself.

Overview

The calculation of the Overview waveform display in the Sample Edit window only takes place after recording.

Just to remind you: the overview calculation always runs in the background, so you can continue to work while it is calculated. Double-clicking on the overview float window opens a dialog box, which can be used to do the following:

break off the calculation (Abort),

carry on calculating (Continue),

or complete the calculation more quickly by bringing it into the foreground (Finish). If you select this option, you will not be able to perform other tasks while the overview is calculated.

Pan

CBX-D3 hardware does not have panning facilities. Individual tracks are hardwired to the outputs with the same number as the track.

Short Regions

Short sections (either regions, or selected areas in the Sample Edit window when the playback monitor and cycle mode are both engaged) should not be shorter than about 7000 samples, if you want them to play back correctly. If you have a badly fragmented or slow drive, this minimum length could be as much as twice as large. Perfect reproduction of sections shorter than this cannot be guaranteed.

8.5 DR 8/16

Logic supports the DR 8 and DR 16 hard disk recording systems by Akai. Please note the following limitations:

The DR is the clock master, meaning Logic Audio Platinum is slaved to MIDI Time Code (MTC). Under **Options > Settings > MIDI Options,** you should set “MIDI Machine Control (MMC)". MIDI and SCSI connections are required.

Logic Audio Platinum saves the current playlist found in the DR. When quitting Logic Audio Platinum, it will prompt you to send back the original playlist found when Logic Audio Platinum was launched. This option can be switched off under **Audio > Audio Hardware & Drivers > DR8/16 > Restore original playlist before quit.**

Logic Audio Platinum is able to display the output levels for each audio track, reading them from the DR through SCSI.

This may cause SCSI noise. You can switch off this function under **Audio > Audio Hardware & Drivers > Request Levelmeter thru SCSI.**

The DR has its own disk(s), the contents of which are invisible to the Mac. For this reason, the DR file references are mirrored on the Mac bootdrive in a folder named "(DR-Files)“. This folder is automatically updated to the current state found in the DR everytime Logic Audio Platinum is launched, so you don’t need to be concerned about it. These “Alias” files are needed to save basic information, and the overview data. The files are small.

Arrange:

Logic Audio Platinum allows you to grab the whole playlist from the DR, and adds all regions to the Arrange in one operation. Select **Audio > Akai DR8 > Get complete playlist from DR8/16 .** This is useful if you have recorded into the DR outside of Logic, and wish to begin working with that recording in Logic. This is an update, existing regions are replaced by the new ones.

The same is possible for one, selected audio track. **(Arrange > Options > Audio Setup > Get playlist of selected track from DR8/16).**

Any new recording is to be handled on the DR itself. When a recording is finished it is easy to grab the new recorded regions using the above described functions.

To update the DR's playlist to the new or changed regions, the whole playlist must be sent from Logic Audio Platinum to the DR. This happens automatically, after a short delay. The DR stops if this happens during playback.

View:

Logic Audio Platinum gets the overview data from the DR for each file. The resolution is ten times smaller then is normally the case in Logic, but it is enough to see what’s going on.

Audio window:

All DR files used in the Arrange are automatically added and displayed in the Audio window as usual.

The Audio window allows all DR Files to be added in one operation, using the menu entry **Audio File > Add all DR8/16 Files. New** regions can be created, edited and added to the arrange as usual.

The DR files can be renamed with the DR limitation of 10 characters. (Note, that sessions created within the DR itself might use these files).

Selected audio files can be deleted using **Audio File > Delete File(s).** This helps to recover disk space within the DR. (Note: this operation can't be undone).

Selected audio files can be copied to a Mac drive by using **Audio File > Copy File(s) > SDII** or AIFF. Note, that this can take a while, due to the SCSI performance of the DR (see the section ***Sample Edit*** below).

Environment:

As with other audio systems, audio objects are created for the DR tracks.

With the current DR firmware, volume and panning changes are sent to the DR thru SCSI. This is slow, but this behavior may be changed when the MIDI Sysex structure is made available by Akai.

Sample Edit:

It is possible to open the Sample editor, but you will be warned (an alert is given the first time): that the read/write performance (speed) is very poor. If you start a process, it can be aborted by typing ⌘.. This behavior is not due to any a problem with the Mac or Logic Audio Platinum, but is a limitation in the DR's architecture.

8.6 1212 I/O

System Requirements

You need a PowerMac, or MacOS compatible computer with PCI expansion slots to run a Korg 1212 I/O. Be sure to consult the manufacturer's documentation to ensure that all equipment is installed and functioning correctly.

Please make sure that the System extension "1212 I/O" is in the folder "Extensions ", inside your System folder Please use version 1.1 (or higher). Ask your dealer or Korg distributor an update, if necessary.

Features

1212 with Logic

Logic Audio Platinum currently supports operation with one 1212 I/O card. This card has 8 inputs and outputs in ADAT Optical Link Format, 2 inputs and outputs in S/P-DIF format, and 2 analog inputs and outputs. All inputs and outputs can be used in parallel.

Audio Tracks

If the 1212 driver is switched on, whenever you create a new song, all the necessary audio objects are created automatically. You can copy this Environment layer into your Autoload Song.

Bit Depth

The Korg 1212 S/PDIF in and outs support 20 Bit-the other inputs and outputs (Analog, ADAT) only deliver a 16 Bit signal. Logic Audio Platinum supports 20 Bit recording and playback, by recording 24 Bit files, the last 4 bits of which are ignored.

De/Reactivating Drivers

If you wish to run Logic without the 1212 Extension, or any other audio drivers-in other words, simply as a MIDI program-do the following:

Open **Audio > Audio Hardware & Drivers**
Deactivate the check box next to "Korg 1212 I/O".
Quit Logic.

The 1212 Extension will be deactivated on the next program start. Less RAM is required when the driver is deactivated. To reactivate any installed driver, simply activate the appropriate check box.

Special Functions

Input Gain

The Korg 1212 I/O allows the regulation of input gain. Open **Audio > Audio Hardware & Drivers,** and select an appropriate value between 0 (off) and 255 (maximum).

Synchronization

In the dialog window **Audio > Audio Hardware & Drivers you** can select between these Word Clock sources: Internal, S/P-DIF, ADAT.

Audio Objects

Input Routing

In every Track Object, you can select the input, by using the top button in the I/O section.

Output Routing

In every Track Object, you can choose the output pair, by using the lower button in the I/O section. Select either the left (odd) or right (even) channel of the output pair, by using the Pan control.

8.7 TDM Hardware

Logic Audio Platinum (not Gold) allows you to make use of the TDM functions supported by ProTools III, ProTools 24 or Mix Plus.

What is TDM?

TDM stands for “Time Division Multiplexing”, the time-interlaced transmission of several digital audio signals through one data bus. This bus system is physically isolated from the computer system bus, and runs between the individual TDM-capable plug-in boards. 256 digital audio signals with a word length of 24 bits each, can be transmitted on the TDM bus, comprising the signal paths within a virtual mixer. These signal paths are necessary to insert plug-ins, which are calculated on the DSP card (DSP = Digital Signal Processor), into the individual channels, or to select them through auxiliary buses, or to direct the 8 digital outputs of a SampleCell II TDM card to the virtual mixer.

Please note, that with the ProTools III hardware, you can only record if the audio hard drive is connected to the disk I/O card. A hard drive connected to the computer’s SCSI bus cannot be used. The disk I/O card has its own, independent SCSI-Bus, even though the “Mount Digi" software included with Digidesign or the “Digidesign System INIT" show the connected disk(s) on the MacOS desktop. The newer ProTools D24 and ProTools Mix cards no longer use their own dedicated SCSI bus, and can use any drive mounted to the System bus.

However, you should always use a drive from the approved list provided by Digidesign.

System Requirements

To operate with TDM support, you will need a Macintosh, Power Macintosh, or a MacOS compatible computer with properly installed ProTools III (PCI or NuBus), ProTools 24 or ProTools Mix audio hardware.

In order to utilize the 24-bit capacity of ProTools 24 or ProTools Mix, Logic Audio Version 3.012 (with an installed TDM Extension), or Logic Audio Platinum 3.5 or higher is needed.

Logic Audio Gold is not compatible with ProTools 24 or ProTools Mix hardware. ProTools III-Hardware, however, can be run with Logic Audio Gold (without TDM capacity- like a Session 8 System).

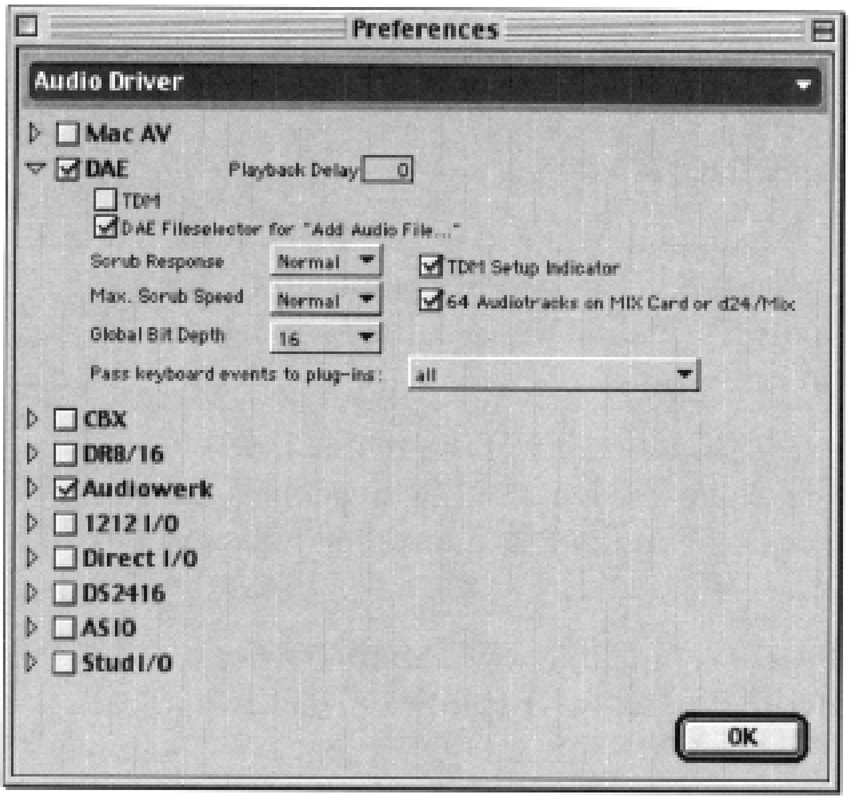
We recommend that you ask your Digidesign dealer to install the ProTools hardware, and to configure your computer.

Turning Your Extensions On and Off

If, for any reason, you should wish to run Logic without its TDM functions, you can switch off the TDM driver at any time.

This facility can be particularly useful if you are booting up Logic with only a small amount of RAM allocated for its use, if you are using a slow computer or if the necessary hardware is temporarily unavailable (in another studio, being repaired, etc.). If you switch off the driver you will also prevent any associated error messages from appearing when you boot up. These always appear if an activated driver cannot find its associated hardware anywhere in your system.

Select **Audio > Audio Hardware & Drivers.**



Click on the check box by the word "TDM" (see above).

The "x" will disappear, and you can now use your ProTools III hardware like a Session 8 system.

You may disable the entire DAE driver, using the same steps, except that you should click on the check box to the left of the word DAE in the line above. After disabling all audio drivers, you can use Logic exlusively as a MIDI program. This is particularily useful if the computer has no audio hardware, of if the existing audio hardware will be controlled by other software during sequencer operation.

Switching the Driver Back On

To turn the TDM or DAE drivers back on, just click again on the same checkboxes. However, before you can make use of TDM or DAE again, you will have to restart Logic.

Object Parameters of TDM Audio Objects

For general information on the audio object parameters, see the section ***The Object Parameter Box*** on page 6-104.

Cha

The Channel parameters on TDM audio objects have a few extra settings:

Bus

When this is selected, the audio object becomes a bus fader, for example when in use as part of a subgroup. You could for instance, route several track objects' outputs to buses 7/8, and then set the ***Bus-7/8*** object to use the physical outputs 1/2. You can then use the ***Bus*** object to adjust the level of the subgroup (7/8), which will be sent to the master output (l/2).

Aux

The ***Aux*** object is a variable (input and output) object. If you use one of these objects to handle the signal from Input 1, this will have the same effect as using an ***Input/ object.***

The advantages of an Aux object are that you can:

- assign various insert effects to several Aux objects, and apply them to Input 1 (whereas an Input-1 object could be used only once),
- change the signal source for the Aux object at any time (all other objects have a fixed signal source; e.g., Track, Input or Bus).

Initially, only Aux 1 and Aux 2 are available in the Selection Menu. Once you have used all of the available Aux objects, Logic Audio will make more available as needed.

You will probably use an ***Aux*** object primarily to set a ***Bus*** as an Input, on which Aux Sends of various audio tracks will be transmitted.

You can also use the **Aux Object** with a bus input as Aux Master Send for an external effects device. Select an output of the audio interface to be used as Aux object output. Then connect the input of the external effects device with this output. Now, you can direct the output of the external effects device back to the virtual mixer.

Editing Plug-ins

Plug-in Key Commands

With Logic 3.0 and later versions, you can control Plug-ins with their own key commands. For example, if you activated an area of the TDM Plug-in with the mouse, you can enter numeric values in that section. To avoid accidental entries, you can click on the background of the Plug-in window, or press the esc key, after you have changed a parameter.

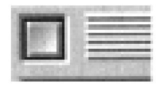
Any key commands that are not used by the Plug-in, will be passed on to Logic. Under normal conditions, this ensures that functions like Start and Stop will continue to work. If this is not the case, you may use several new options to define that only certain key combinations will be passed to the Plug-in, while all other key presses on the computer keyboard will be received by Logic. These options are located in **Audio > Audio Hardware & Drivers,** in the DAE section ("Pass keyboard events...").

Plug-in Setups (Settings, Effect Programs)

All TDM Plug-in settings are saved with the song, and will be automatically restored when the song is loaded.

You can also use Plug-in Setups as a practical means to manage the parameters of a Plug-in. This method is comparable to the storage locations of an external effects device, combined with a clipboard for effects parameters.

In the header line of the Plug-in window, you can open the Setups pull-down menu by clicking on the arrow button:



Copy Setup copies all parameters onto its own independent Plug-in clipboard. They remain here, until the next time you select Copy *Setup*. This does not affect the global Logic clipboard.

Paste Setup If you have opened a plug-in of the same type, you can paste the parameter set from the plug-in clipboard. This allows you to quickly set up several plug-ins of the same type, or exchange effects settings between songs.

Save Setup This allows you to save all the plug-in parameters to disk. This is useful if you have created a special sound effect, which you want to have available for future use.

***Load** Setup* For loading stored parameter sets from disk. The file selector box shows only setups for compatible plug-in types. It can also read the Digidesign format for TDM plug-in settings.

The parameter setups of a plug-in can also be copied between the plug-in's mono and stereo versions, and the Digidesign format for plug-in Settings can be read (the supplied effects programs of most plug-ins are saved in this format). When loading Setups, through the arrow menu in the plug-in windows, Logic automatically displays the corresponding files in the file selection box. Settings files are usually located in a "Settings" folder, within the DAE folder. The name of the last preset chosen will be indicated by a dot in the menu listing.

DAE Plug-in Folders

Starting with DAE 3.1, the DAE folder contains only one "Plug-ins" folder which can store both AudioSuite and TDM Plug-ins.

Disabled Plug-ins

If you haven't done so, create a folder with the name "Plug-ins (unused)" in your DAE folder (or in any other location). Then move all TDM Plug-ins not used on a daily basis, into this folder. This will speed up the DAE (and Logic) boot process.

Support of TDM Plug-in Sidechains

Some TDM Plug-ins, such as Noise Gates or compressors, allow Sidechain Signals from other Tracks or Busses. This option is available in Logic Audio.

Automation

For detailed information on this subject, read the chapter Chapter 6: ***Mixers, Effects and Audio Objects.***

As with other audio objects, TDM objects can be automated, although only adjustments made to the controls on the first four plug-ins can be recorded. Up to 16 parameters may be remote-controlled on each of these four plug-ins.

A fixed hierarchy has been adopted for the controller numbers used to automate plug-in parameters, based on the position of the plug-in. Plug-ins are counted from the top downwards.

1. Plug In	Controller # 64 to # 79
2. Plug In	Controller # 80 to # 95
3. Plug In	Controller # 96 to #111
4. Plug In	Controller #112 to #127

Dynamic Controller Assignment

To automate the parameters of plug-ins with more than 16 parameters (e.g., D-Verb, DPP1 or JVP), Logic Audio manages the number of required controllers dynamically.

The "base addresses" (the first controller numbers: 64/80/96/112) remain unchanged. However, if you have activated only

one plug-in, in the first position (first slot), you can automate 64 parameters in this location: 64-127.

If you insert an additional plug-in on the second slot, controllers

64-79 (16 parameters) will be available for the first plug-in, and

80-127 (48 parameters) will be available for the second plug-in.

If you use one plug-in in the first slot and one plug-in in the third slot, 32 controllers (64-95 and 96-127) will be available for each of the two plug-ins.

To find out which controller number controls which parameter for a given plug-in, open the Event List and click on a controller to select it. A list of all parameters that can be automated will appear.

In the Event List, the parameter names of all known plug-ins are shown in plain text.

Caution: Not all TDM Plug-ins (in particular older plug-ins) allow for problem-free real-time automation of parameters. In such cases, the basic settings still can be saved together with the song, however, the dynamic automation of the parameters during song playback is not possible.

Automation Process

The automation of plug-in parameters works the same as the regular automation described in the manual.

Just try it yourself. Select a track assigned to the ***A-playback*** instrument, and start recording. Open a plug-in, on an audio object insert, and move an operating control of the plug-in. Your moves will be recorded as MIDI commands, which can be recalled and edited.

This procedure can be used to detect which controller numbers can be automated graphically with Hyperdraw Using the Auto Define Hyper-

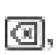
draw function (which also is available as a key command), Hyper-draw will set itself automatically for the first used controller.

Some plug-ins, like the Digidesign Plug-in "Dynamics", respond to the controllers they receive "from the outside", but they will not automatically update the operating control display. Therefore, Logic Audio will redraw the complete plug-in after a certain period.

Plug-ins in objects like "Input", "Aux", "Output", or "Bus" can be automated as follows: Create a channel splitter in the environment **New > Channel Splitter),** and use cables to connect its channels with the respective audio objects. Insert the channel splitter into the Arrange window just like a track instrument (the same way you inserted the ***A-playback*** instrument earlier). Name the channel splitter something like ***"Bus Playback", so*** that you can distinguish it from the existing ***A-playback*** instrument.

Functions, Special Features

Deleting Objects

If you delete an audio object in the environment, by selecting it and pressing , the level of this object will be set to "Zero" (minus infinite dB). However, "Output" or "Bus" objects are an exception: When deleting these, the level will be set to 90 (0 db); otherwise, the tracks routed to this output would not be audible any longer. The other settings (send paths, plug-ins, and sound control) are not deleted but merely muted. This way, Logic Audio automatically ensures that DSP processor capacity will not be wasted. As soon as the object is displayed again in the environment, the last selected level will automatically be restored. In the TDM System, the Send, and Plug-in will be restored.

Special Features of ProTools III with 32 or 48 Tracks

Logic Audio supports 32 or 48 track ProTools III systems, with more than one Disk I/O card, and 64 tracks with the ProTools Mix system.. Please remember the following, when operating a ProTools III system with more than 16 tracks:

Every Disk I/O card has its own SCSI bus. This means that tracks 1-16 can only play back audio files from hard drives which are connected to card 1, tracks 17-23 only from disks connected to card 2 etc. The same applies to recording. The recording path of every Disk I/O card is managed separately in the Audio window through the dialog box **Audio File > Set Record Path...**

Logic will notify you in the Audio or Sample Edit window if you accidentally try to playback an audio file on a track from another card. Such "erroneously assigned" audio files will not play in the Arrange window.

Since MIDI cannot differentiate more than 16 channels, you will need additional Channel Splitter objects ("A-Playback" instruments) for the automation of more than 16 tracks on any TDM system. If you want to create a completely new song (instead of opening a copy of your Autoload Song), e.g., with alt, Logic will automatically create the following Environment Setup:

Tracks 1-16 are connected with channels 1-16 of the Channel Splitter *A-Playback*,

Tracks 17-32 are connected with channels 1-16 of the Channel Splitter *B-Playback*

Tracks 33-48 are connected with channels 1-16 of the Channel Splitter *C-Playback*

(Initially "A-Playback" stood for "Audio-Playback". If you don't like the new name, you can easily rename the channel splitter objects.)

If you operate more than one audio interface (through additional Bridge I/O cards), the additional interfaces are called "b", "c", etc.; for example, “Out 882b 1-2".

Special Features of ProTools 24

The most significant new features of PtoTools 24 are:

Under Logic you now can also record and playback with 24 bit word length.

Provides up to 32 audio tracks. For system requirements, please see the Digidesign documentation.

A core system may have 24 inputs and outputs (versus 16 with ProTools III PCI, or 8 with ProTools III NuBus).

ProTools 24 does not have its own SCSI bus. It uses the computer’s SCSI bus. Therefore, you should read QT movies and audio data not only from separate hard disks, but also from hard disks connected to different SCSI busses.

For the operation of ProTools 24 with TDM, Logic Audio Platinum is required. Unlike ProTools III, it cannot be run as if it were a Project or Session8 system, with Logic Audio Gold. However, you can run a ProTools 24 system in Logic Audio Gold, using the Direct I/O driver. This gives you no access to TDM functionality, so if you have such a system, it is highly recommended that you upgrade to Logic Audio Platinum.

TDM Mixer Plug-in

Please be sure that there is only one Mixer Plug-in in the “Plug-ins” folder (within the DAE folder), which will be either the 16-bit version or the 24-bit version.

24-bit files can also be played back through the “16-bit optimized mixer” (or vise versa). However, only the “24-bit optimized mixer” can adequately handle the resolution of recordings in 24-bit format.

The 24-bit mixer requires more DSP capacity than the 16-bit mixer. Therefore, you may get a DAE error message ("DSPs maxed out”), when opening older songs with maximum DSP load, after the upgrade to ProTools24. In this case, you should use the 16-bit mixer plug-in.

Changing Global Bit Depth

If ProTools 24 hardware is being used, Logic will ask at the initial program start, whether you want to use 16-bit or 24-bit word length. Later, you may change the “Global Bit Depth” in the dialog box **Audio > Audio Hardware & Drivers** from 16-bit to 24-bit, and vice versa.

In most cases, aresolution of 16-bit is adequate for the individual tracks of a regular multi-track production, This setting requires less space on the storage medium, and also less data throughput capacity on the hard disk, to play the same number of tracks. The 24-bit format is mainly advantageous with highly dynamic material (e.g. classical music productions),

Program Start

If you are loading a song with different word length audio files, Logic will display a warning. You can change the word length (see above), or you may convert the audio files.

Please note that the conversion of 16-bit recordings to 24-bit recordings does not bring any sound quality improvement, while a conversion from 24-bit to 16-bit results in the irrevocable loss of the added dynamics possible with 24-bit files.

Converting the Word Length

In the Audio window, select all of the files to be converted.

For the selection, it would be useful to sort the audio files by their word length (bit depth) (see above). Then select all files to be used in the arrangement with [Edit > Select used].

Select **Audio File > Copy File(s).**

In the following dialog box, you have the option to set the word length (bit depth).

After confirming the dialog, the audio files will be copied-the copied files will have the selected word length.

Logic now will ask you if the references in the current song should be changed to reflect the copied files, instead of the original files.

If you want to work immediately with the current song, you should confirm this dialog.

Save the song with a different name by selecting **File > Save as...** if you want to keep the references to the original files in the original song.

Sample Editor

All functions of the Sample Editor can be used with 24-bit audio files. You can even exchange sound material in either direction between 16-bit files, and 24-bit files by using the Copy and Paste commands.

Please note that Premiere Plug-ins use only 16-bit resolution (you can still edit 24-bit files). AudioSuite Plug-ins support 24 bits.

Simultaneous Operation of Different Hardware

Please note the following, if you want to use a TDM System and other audio hardware at the same time.

Generate Audio Mixer

New Song

With the selection of the command **File > New** (or ⌘N), a copy of your Autoload Song will be opened. If you want to open a completely new song with standard defaults (Default Song), just press ⌥⌘N.

Corresponding Audio Objects

Based on the connected hardware, and the installed extensions, this song will include audio objects that correspond to the currently active audio hardware, in the Audio layer of the Environment. It also includes the audio playback instrument, which you will set as the current track instrument for the recording of fader movements.

You may select the complete layer (⌘A), copy it to the Clipboard (⌘C), and then insert it into a new layer in your Autoload Song (⌘V). Or, you can drag the required objects to the environment of your Autoload Song.

If you want to integrate the new “TDM Mixer” into your existing Audio layer, make sure that the position of the audio objects will be maintained during the copy process. To avoid new objects being placed on top of the existing ones, select the existing objects prior to inserting the new ones and drag them to the side or to the bottom (“grab” the object at its right edge, at the object name or with the •J key). Don’t forget to deselect all objects by clicking on the background; otherwise all selected objects will be replaced with the inserted objects.

Finally, don’t forget to save your new Autoload Song (⌘S).

Control Playback

To control playback in the Audio window or in the Sample Edit window, make sure to select the correct corresponding hardware and channel, under Channel and Device (at the left side of the window).

For example, it is not possible to play back an audio file located on a hard disk on the SCSI bus of a ProTools III System through the internal Mac hardware, since only it can access the SCSI bus of the I/O card.

In order to play such a file, the audio file would have to be physically copied or moved to a disk on the system SCSI bus (using the Copy or Move command).

Total Number of Audio Tracks

When operating different audio hardware simultaneously, the number of possible tracks cannot be determined by merely adding them up. This is especially true if the connected systems have to use the CPU processor's power. This is the case primarily in AV mode, but also when using DAE.

The simultaneous operation of DAE hardware and AV audio tracks will result in a significant reduction in the number of usable AV tracks. However, stand-alone systems, like the Akai DR-8/16, cause only a minor load increase on the CPU.

The arithmetically added number of tracks may also be reduced, if the connected systems access the same SCSI bus. This is the case, for example, when using AV tracks and CBX hardware at the same time.

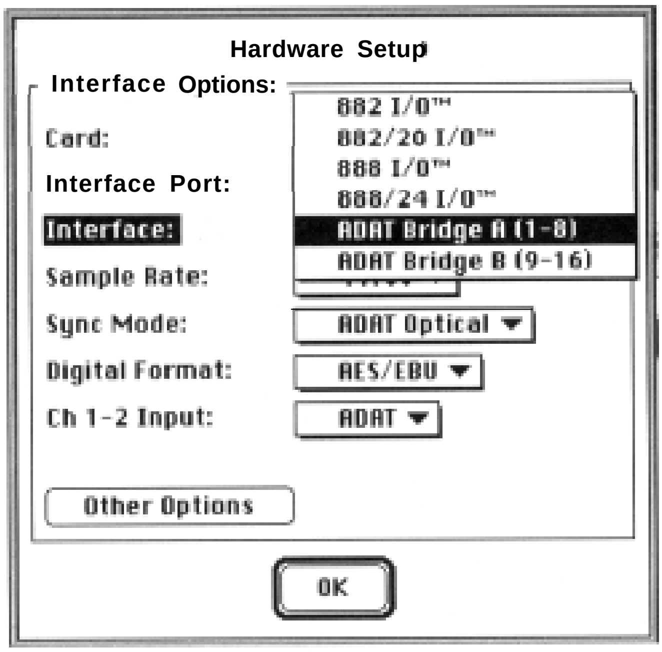
This effect may be significantly reduced (i.e., the total number of tracks increased) by using different hard drives for the various systems (e.g., AV reads from an internal drive, and CBX from an external drive).

On the other hand, the simultaneous operation of ProTools III and CBX (=4 tracks or 8 tracks with two CBXs) presents no problem at all.

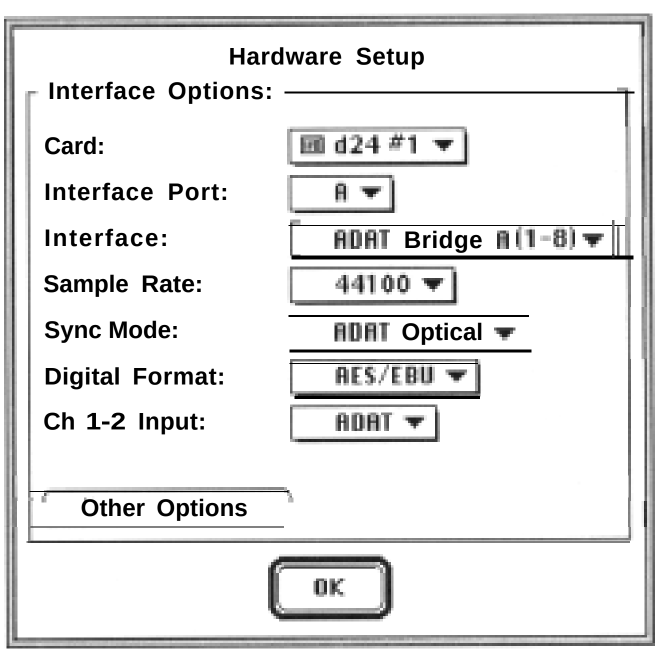
We recommend, however, that you avoid the use of an internal hard disk for hard disk recording, if possible.

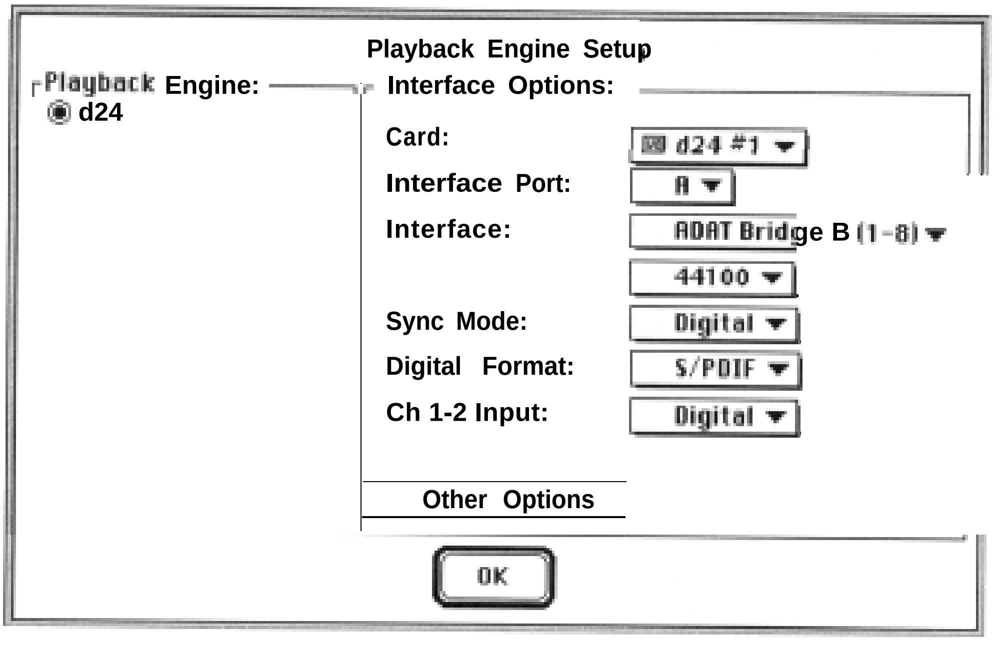
Digidesign Adat Bridge Interface

If you want to use the Digidesign Adat Bridge Interface, you need DAE Version 3.2.3 (or higher). The Adat Bridge Interface is a 19" unit with two optical Adat inputs and outputs providing 16 channels, where channels 1 and 2 also act as AES/EBU and S/PDIF ports.



Each of the two groups of up to eight channels can be individually connected to a Disk I/O or a PCI DSP Farm. The PT24 Disk I/O has two interface sockets. You can select the bridge interface in the **Hardware Setup** menu.





8.8 Direct I/O

Until now, Digidesign hardware has always been addressed via DAE (Digidesign Audio Engine). Digidesign has now released hardware like “Project II”, that is not supported by DAE. The new Audio Engine has been developed to use these cards with Logic Audio. As the Direct I/O interface supports other Digidesign hardware as well, the driver is called “Direct I/O” instead of “Project II”.

Please note that the system extension Digisystem INIT 3.4 or higher is required. This system extension is shipped with the card, but is also available from the Digidesign distributor in your country, or from the Digidesign web site www.digidesign.com).

By the way: users of other Digidesign PCI cards, (e.g. Audiomedia III or ProTools Project) may also benefit from Direct I/O. You achieve a higher number of tracks (if your CPU is fast enough), and gain access to Logic Audio’s internal effects and VST Plug-ins. Even if you are using a ProTools III Core system, you may use Direct I/O, at least if you have a powerful CPU. However, please note that no TDM functions will be

available. Furthermore, you should only play back audio files from the hard drive attached to SCSI bus of your Power Macintosh, and NOT from any drive attached to the SCSI bus of the Digidesign card when using Direct I/O. Newer and larger ProTools 24 systems can be also be run via Direct I/O, instead of DAE. However, as you then lose all TDM functions, it does not really make sense.

In order to activate the audio driver for Digidesign Project II, or other hardware, go to the main menu and select **Audio > Audio Hardware & Drivers > Direct I/O** . Here are descriptions of the parameters you will find in the “Direct I/O” section for the new Audio Engine:

I/O Buffer Size

This parameter, which affects Direct I/O, governs the latency and the performance of the card. A value of "1" is an ideal setting for good performance and reasonable fader response. Smaller values reduce the monitoring delay and the fader response time, but also reduce the number of audio tracks and the performance of plug-ins.

Use 16 Ins & Outs

The Project II card allows you to use two audio interfaces, with 8 inputs and outputs each. If you have only one interface, please switch this setting off. This saves memory, and optimizes overall performance. You should only switch this setting on when actually using two interfaces.

Digidesign Hardware Setup

In the audio window, with **Options > Digidesign Hardware Setup**, you can open a dialog that allows you to select the audio interface, and to switch between analog and digital input. The equivalent dialog box for setting up DAE is only available when actually using DAE to run your Digidesign hardware. The Direct I/O driver cannot be used simultaneously with DAE.

8.9 Sonorus StudI/O

In comparison with the ASIO driver, this direct support of the StudI/O offers the following advantages:

Better performance
More precise start when recording and playing back.

Operating Modes

The StudI/O can be operated in 5 different modes. All modes allow 24 bit:

2x ADAT

16 In/Output channelsalways available in ADAT format.

ADAT + SPDIF

8 I/O channels in ADAT format, and two channels of S/PDIF I/O.

ADAT + BNC

8 I/O channels in ADAT format, and ADAT-sync

2x SPDIF

Two twin channel S/PDIF I/O.

96K SMUX

8 I/O channels over ADAT with 88.2 or 96 kHz format (Platinum only).

With regards the other parameters that appear here, please refer to the manual of the StudI/O

8.10 ASIO

ASIO (Audio Stream In/Out) is a common driver format introduced by Steinberg to allow audio hardware and audio software

(in this case Logic Audio Platinum) to work together. An ASIO driver translates the input data of the hardware into a specific format, and transfers the data to the software (Logic Audio Platinum), which receives it as recording data. In turn, the software (Logic Audio Platinum) transmits the final mixed signals to the ASIO driver, which then transfers it to the hardware.

The ASIO driver is generally provided by the manufacturer of the audio hardware.

The advantage of this sort of industry standard interface is:

New Audio hardware is compatible with Logic Audio right away, as long as it is equipped with an ASIO driver.

The disadvantage of this sort of industry standard interface is:

To software each driver (and the hardware supported with it) looks identical. There is no flexibility to support specific features of the hardware. Some manufacturers ship their own control panels, the settings of which are "invisible" to Logic Audio.

Just like Direct I/O, ASIO support is handled by the Audio Engine as described in the section Audio ***Engine*** on page 8-5. The settings for the audio engine and for the ASIO driver can be found in the **Audio > Audio Hardware > Drivers > ASIO** menu.

Current Driver

This is where you select the ASIO driver you would like to use. When changing the Audio driver, it is necessary to quit Logic and open it again.

Logic expects to find the ASIO driver in a folder named "ASIO Drivers", in the directory of the Logic program file.

! This also can be an alias to an ASIO folder somewhere else

ControlPanel

This switch allows you open the control panel of the active ASIO driver, if one is available. Whether or not this control

panel exists depends on the manufacturer of the hardware being supported.

Clock Source

This setting allows you to use a digital clock source from an external device, if the hardware and the driver you are using supports this.

ASIO Buffer Delay

Some ASIO drivers give incorrect information regarding input and output delay. The difference from the actual value may vary between different versions of the driver, therefore it is necessary to allow changes of this setting. Please check our recommendations for specific hardware systems in the ReadMe-File on the CD-ROM.

Max. I/O Streams

This parameter is used to define a maximum of inputs and outputs. This is needed especially for the DS2416 card, which does not run with its 8 ins & 16 outs on machines slower then a G3/333. The default is set to 4 ins and 8 outs for the DS2416 For others it is set to ***Max***, which means that you can use all ins and outs.

It's recommended that you switch off unused I/O channels in the driver (control panel), if you are using an ASIO driver that supplies one.

System Memory Requirement

Changing some of the above settings increases or decreases the required memory. The required memory is displayed on the right side of the "ASIO" switch.

The Key Command ***Open System Performance...*** will open a window that displays the CPU and Hard Disk Bus usage as a percentage (with 100% being the maximum capacity).

8.11 Yamaha DSP Factory

The currently supported components of the Yamaha DSP Factory System include one DS2416 card and, optionally, up to two AX44 extension bays (4 analog ins and outs each) or an AX16T (2 x 8 ADAT Ins and Outs). Furthermore, Logic Audio Platinum is also ready to support up to 8 channels of one SW 1000XG soundcard and future I/O extensions whose specifications are yet unknown. In order to use the card, select **Audio > Audio Hardware & Drivers** and check DS 2416.

Features

(If you use one DS2416 only)

Up to 16 audio tracks

24 channel digital mixing disk, including

- up to 4 channels for two stereo effect returns of the two internal multi-effect processors
- up to 4 Input channels. Further inputs are only available with the AX44 hardware extensions and by sacrificing audio tracks (there is a total limit of 24 channels).
- Variable output routing for the stereo sum and the 4 Aux sends. In addition, the two internal effects sends can be used in parallel. This results in a maximum of 6 individual outputs.

Each of the 24 channels offers:

- 4 Equalizer Plug-ins (fully parametric, shelving)
- Dynamics Plug-in (Compressor, Limiter, Expander, Gate)
- Channel Delay

In addition, the two effect return channels each offer:

- One Multi-Effect Plug-in (choose from a list of 40 algorithms)

Mixer

The mixer directly reflects the available DSP and routing of the DSP Factory, as long as you use the ASIO Driver DS 2416 as the only driver. It is defined by the feature set of the hardware, and therefore is not completely variable. Please note that Logic does not utilize its own audio driver, but uses Yamaha's ASIO driver, and the real-time effects of the digital mixer integrated in the DSP Factory.

The default configuration of the mixer offers 16 audio tracks, 4 inputs (analog and digital input), two stereo effect returns and two stereo output objects (master fader) for the stereo analog and digital output of the DS 2416 card. The mixer offers 24 channels. Additional inputs always reduce the number of audio tracks.

Fader

The faders have a different range than the faders which control other hardware: 0 dB equals the volume controller value 127. The value 90 equals -6dB. Unlike other audio drivers, the fader for the DSP Factory can only decrease the volume, but not increase it.

Inserts

Each mixer channel holds up to 4 EQs, one dynamic module, one channel delay and one phase inverter.

EQs, dynamics modules, channel delay, and the phase inverter are used as inserts and can be controlled via plug-in windows. The actual signal path is not variable and goes like this: EQs -› Dynamics -› Delay.

The plug-ins of the audio tracks are used-as is common-on playback. However, please note that effects on the input channels (i.e. channel 17 to 20 in the standard configuration) are also recorded. For unaffected recording you might want to set the inserts of the input channels to bypass. (⌥ ctrl-Click). On the

other hand, it might be useful to directly record EQs and dynamics.

On the effect return channel, the internal multi-effects devices always use the first insert slot, but EQs and dynamics can be used in the following ones.

Sends

Track channels (audio tracks) and input channels have up to 8 sends. The sends labeled “Effect 1" and “Effect 2" address the two internal effect busses.

In addition, the sends of the available outputs of the installed hardware can also be used; e.g. output 1-4, if one DS2416 card is installed (using an AX44/AX16 extends the choice to outputs 1-6). Please also refer to the following section on input/output routing.

As is common with Logic’s fader, the effect sends are processed after the fader (post-fader), but can be switched to pre-fader using the flip menu of the active sends.

Input/Output Routing

Located above the pan fader is the output routing for all 24 channels. This allows you to direct the output signal of a track to one of the outputs. The track channels (“Audio 1-16") also feature input routing; i.e. the ability to select from which input you would like to record.

The number of available inputs and outputs depends on whether you have only a DS2416 card installed, or whether you are also using the AX44 I/O box (or AX 16):

DS2416 4 inputs and 4 outputs

DS2416 + AX44: 8 inputs and 6 outputs (the two additional outputs can only used parallel to the internal effect busses).

DS2416 + AX16: 8 Inputs and 6 Outputs (2 more Outputs can be used in parallel with the internal effect busses; Recording busses are restricted to a maximum of 8 Inputs.

A second AX44 may be used but does not increase the number of outputs you can address simultaneously (max. 8)

Default:

Input 1 = Channel 17 = Analog In L

Input 2 = Channel 18 = Analog In R

Input 3 = Channel 19 = Digital In L

Input 4 = Channel 20 = Digital In R

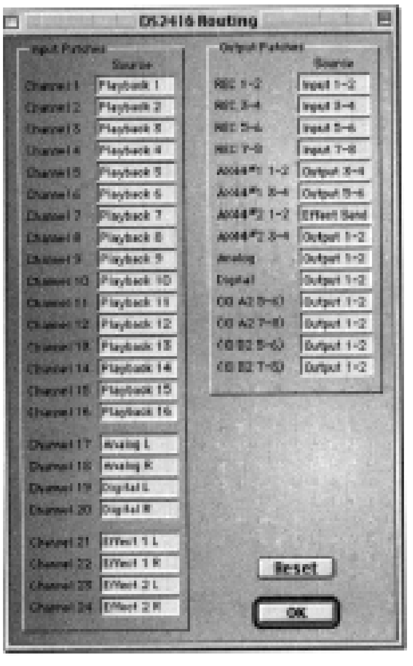
Both the Analog Out and the Digital Out are addressed by "Output 1-2".

The first AX44 (if installed) is addressed by “Output 3-4” and “Output 5-6".

The second AX44 (installed) mirrors effect sends 1 and 2 on its first output pair.

These default settings can be edited in the "DS2416 Routing” window.

DS2416 Routing



You can open this window with a key command or from the audio window **Options > DS2416 Routing**.The left half is a sort of digital patchbay for selecting the inputs of the mixer channels 1-24. The right half is basically a patchbay for the outputs of the hardware.

The input patches can be configured with certain limitations; e.g. the 4 inputs of the first AX44 can only by inserted on channels 17-20 (thereby sacrificing the inputs of the DS2416) or on channels 13-16 (thereby sacrificing 4 audio tracks). The inputs of the second AX44 can only be inserted on channels 9 to 12 (losing another 4 tracks). The 8 outputs of the SW1000XG soundcard ("Sub 1-8") can only be inserted via channels 9-16 or 17-24.

Please note that this window already displays some options which are not yet available as hardware, e.g. the hardware for the options "IO-A2" and "IO-B2". Logic Audio should run with this hardware, although we were not yet able to test it as of the release of Version 4.0.

Clock Source

The parameter ***Clock Source*** under **Audio > Audio Hardware > Drivers > ASIO** allows you to switch between the use of internal synchronisation or an external digital synchronisation signal. The setting ***serial in*** is needed in case you use a DS 2416 together with a SW1000XG.

If you select ***Digital*** Logic Audio will continously check the sample rate of the incoming S/P DIF signal. If it detects a difference from the currently set sample rate **(Sample Rate** in the **Audio** menu) a warning message will appear. If you confirm

this message with USE, the internal sample rate will be set to the detected rate.